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# (54) A method and a device for providing improved speech intelligibility

(57) The invention relates to a method for reduction of noise in an audio signal containing noise and a target signal, the method comprising, providing at least two input signals; processing the input signals by means of an independent component analysis, hereby determining statistical dependencies of signal elements of the two input signals and determining whether statistical dependent signal elements form part of the target signal; outputting a part of the audio signal. The invention further relates to a device for use in reducing noise in an audio signal containing noise and a target signal.

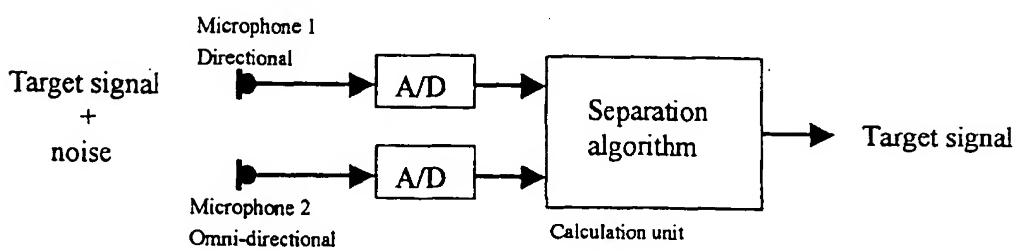


FIG. 2

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**[0009]** The objective of ICA is to recover the underlying independent source signals given only sensor observations that are linear mixtures of the original source signals. The only assumption of ICA is that the original source signals are statistically independent, otherwise the statistics of the source signals and the mixing of these into the sensor signals may be unknown. In contrast to correlation-based transformations such as Principal Component Analysis (PCA, I.T. Jolliffe, Principal Component Analysis, 1986, Springer Verlag), which decorrelates signals according to 2nd-order statistics, ICA also reduces higher-order statistical dependencies, in terms of maximising joint output entropy, in order to extract statistically independent signal components.

**[0010]** In the linear blind signal separation problem, $N$ signals,

$$\mathbf{s}(t) = [s_1(t),\ldots,s_N(t)]^T,$$

are mixed so that an array of $N$ sensors picks up a set of signals

$$\mathbf{x}(t) = [x_1(t),\ldots,x_N(t)]^T,$$

each of which has been mixed, delayed and filtered as follows

$$x_i(t) = \sum_{j=1}^{N}\sum_{k=0}^{M-1} a_{ijk}\, s_j(t - D_{ij} - k)$$

where $D_{ij}$ are entries in a matrix of delays and $\mathbf{a}_{ij}$ are the $M$ - tap filter coefficients between the $j$th source and the $i$th sensor. The problem is to invert this scrambling without knowledge of it, thus recovering the original signals $\mathbf{s}$(t) given only the $\mathbf{x}$(t) signals. Finding this inverse scrambling is a challenging task since no informations are provided about the mixing nor the signals (hence the term blind separation). The type of architecture chosen for inverting the scrambling is important and can be made in numerous ways. An accurate architecture for inverting a M-tap filter is an infinite impulse response (IIR) filter with M coefficients. However, IIR filters are limited to have poles inside the unit circle, which imply that a stable filter only exists for a minimum phase system. FIR filters may be used to approximate the inverse solution. Thus the inverse scrambling is performed according to

$$u_i(t) = \sum_{j=1}^{N}\sum_{k=0}^{M-1} w_{ijk}\, x_j(t - d_{ij} - k)$$

which has filters, $\mathbf{w}_{ij}$, and delays $d_{ij}$, which supposedly reproduce, at the output $\mathbf{u}(t)$, the original uncorrupted source signals, $\mathbf{s}(t)$, apart from a scaling factor for each signal and a permutation of signals.

**[0011]** Several algorithms have been proposed for the blind separation of linear mixtures. Bell and Sejnowski (3) proposed to learn the separating process by minimising the mutual information between components of $\mathbf{y}(t) = g(\mathbf{u}(t))$, where $g$ is a non-linear function approximating the cumulative probability density function of the sources. They showed that for positively kurtotic signals (like speech) minimising the mutual information between components of $\mathbf{y}(t)$ is equal to maximising the entropy of $\mathbf{y}(t)$, which can be written as

$$H(\mathbf{y}) = -E[\ln(f_y(\mathbf{y}))],$$

where $f_y(\mathbf{y})$ denotes the probability density function of $\mathbf{y}(t)$. Denoting the determinant of the Jacobian of the whole unmixing process by $|J|$, $f_y(\mathbf{y})$ can be written as $f_x(\mathbf{x})/|J|$ (the Jacobian is a matrix with entries of $\partial y_i/\partial x_j$). Maximising the entropy of the output leads to maximising

$$E[\ln(|J|)],$$

which in turn can be developed into a stochastic gradient ascent rule using instances of $\mathbf{x}(t)$ and $\mathbf{y}(t)$, instead of using the expectation. Thus

$$\Delta\mathbf{W} \propto (1 - 2\mathbf{y}(t))\mathbf{x}(t)^T + [\mathbf{W}^T]^{-1}$$

where $g(u) = 1/(1 + e^{-u})$ is used to approximate the cdf.

**[0012]** The algorithm can be made more efficient and independent of the conditioning of the mixing process (matrix) by using the so-called natural gradient instead of the absolute gradient, see Amari (11).

**[0013]** One particular proposed application of ICA is within electroencephalographic (EEG) recording of scalp potentials in humans and related brain activity

put signals according to a predetermined scheme.

**[0027]** Due to the fact that most hearing impaired have a hearing disorder which makes it even more difficult than for normal hearing persons to separate a target signal from the noise, which is often present in a speech situation, the invention is particularly relevant in connection with the technical field of hearing aids.

**[0028]** The invention therefor further relates to a hearing aid comprising: at least two microphones for audio signal input; signal processing means in connection with the microphones; an amplifier in connection with the signal processing means; a receiver in connection with the amplifier for outputting a signal from the amplifier; the signal processing means being adapted to process the signals by means of an independent component analysis method or a similar method based on the input from the at least two microphones, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal.

**[0029]** The hearing aid according to the invention may further comprise the features set forth above, either separate or in combination.

**[0030]** Other fields of relevant use of the invention may be telecommunication or audio systems. In such systems the input and output may be connected to antennas or similar transmission and receiving means or may comprise microphones as input means as in the case of a hearing aid. Other elements of such systems may be standard elements, as these are not influenced by the signal processing according to the invention.

**[0031]** The invention will be described more detailed with reference to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

**[0032]**

- Fig. 1 is a diagram showing the principles of the invention;
- Fig. 2 is a schematic diagram showing the principles of an implemented version of the invention;
- Fig. 3 is a schematic diagram showing the principles of an implemented version of the invention;
- Fig. 4 is a schematic diagram showing the implemented version of the invention as shown in fig. 2 further implemented in a hearing aid.

DESCRIPTION OF THE PREFERRED EMBODIMENT

**[0033]** The fundamental principle of the invention is schematically shown in fig. 1. The invention is basically a system, e.g. a hearing aid, with two or more sensors and a calculation unit. The calculation unit carries out the separation of the target and noise signals, by using the independence of the mixed signals according to ICA or a similar method comprising the basics of the ICA. The sensors are arranged so that one is positioned to receive sound primarily from a target direction in front, whereas the others have arbitrary characteristics that do not specifically favour the target direction. Hereby, it is possible to use the technique of independent component analysis to separate the desired signal, which impinges from the target direction, from the disturbing noise signals, which impinge from any other directions.

**[0034]** To illustrate the invention, an example is given of a system implementing signal processing as described above. Fig. 2 schematically shows the signal processing system. The system comprises a directional and an omni-directional microphone, and a digital signal processing unit implementing the signal separation algorithm. Using the directional microphone gives the target direction from in front of the user, whereas the omni-directional microphone gives a signal equally representing all signals around the head of the user.

**[0035]** A particularly important property of the independent component analysis is that it separates convolved and delayed source signals, where each independent source signal is defined as a signal which appears in the same way within each mixing process. Another important characteristic about the independent component analysis is that knowledge about the ratio of the source signals within the mixed signals can be used for classifying the separated signals. If for instance one source signal appears with a significantly better signal to noise ratio in one of the sensor signals, this information can be used to ensure that this source signal always will appear in a fixed output. Within the present invention these two characteristics combined with an appropriate placement of at least two sensors are exploited to eliminate signals not coming from in front of the user of the device.

**[0036]** From fig. 3 an embodiment appears, which comprises the features of the embodiment of fig. 2, but where the signal processor produces two output signals. By means of switching means one of the two output signals may be selected for further processing, e.g. amplification, or for output.

**[0037]** From fig. 4 an embodiment appears as a hearing aid according to the invention. The essential components of the hearing aid comprise two microphones, preferably a directional microphone and an omni-directional microphone, and an A/D converter connected to each of the microphones. The A/D converters are connected to a digital signal processor, which is adapted to perform the ICA method on the incoming signals. The signal from the signal processor is then lead to an amplifier and from this through a D/A converter to a receiver for performing the output of the processed signal. The devices of the figs. is in a usual manner powered by means of usual power sources, such as batteries.

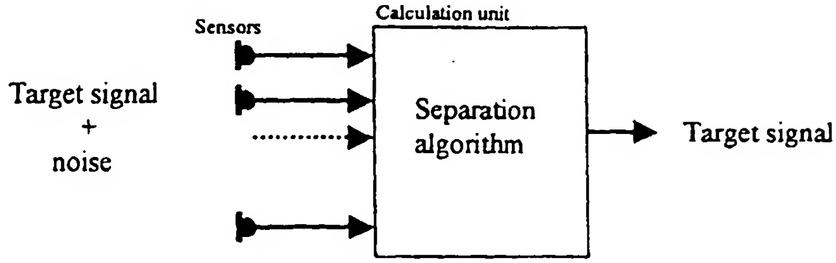


FIG. 1

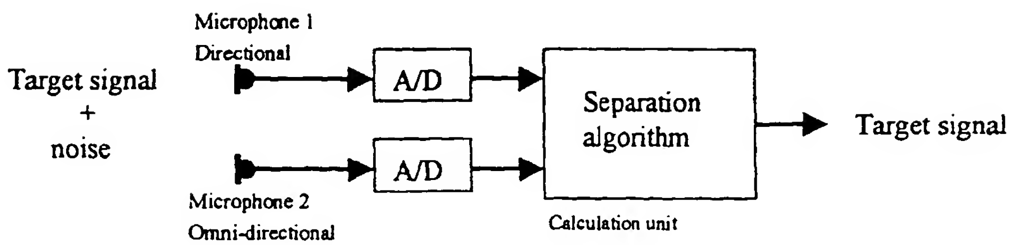


FIG. 2

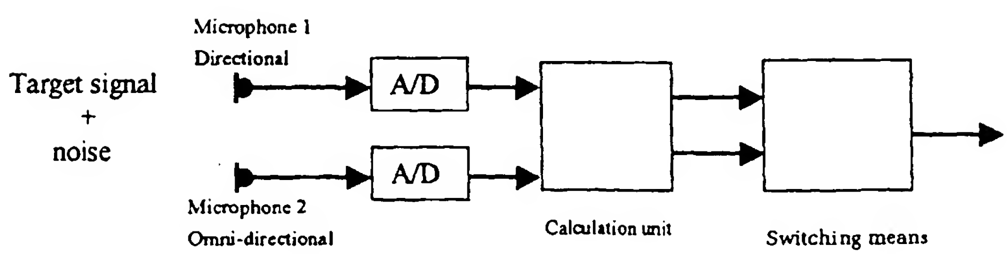


FIG. 3

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# EUROPEAN SEARCH REPORT

Application Number
EP 99 61 0002

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The present search report has been drawn up for all claims

| Place of search | Date of completion of the search | Examiner |
|---|---|---|
| THE HAGUE | 12 July 1999 | Gastaldi, G |

CATEGORY OF CITED DOCUMENTS

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## ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 99 61 0002

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